APPENDIX C

(CLEAN VERSION OF ALL PENDING CLAIMS)

(Serial No. 09/821,256)

CLAIMS

What is claimed is:

1. (Amended) A method for Voice over Internet Protocol (VoIP) telephone calling over an IP-based packet network comprising:

initiating a telephone call to a destination associated with a destination telephone number;

connecting said telephone call to an originating VoIP gateway switch over a public switched telephone network (PSTN);

determining a preferred route from said originating VoIP gateway switch to said destination through said IP-based packet network and through a terminating VoIP gateway switch nearest said destination using enhanced SS7 signaling packets; and

setting up two-way communication through said preferred route using said enhanced SS7 signaling packets over said IP-based packet network.

2. (Amended) The method of claim 1, wherein connecting said telephone call to said originating VoIP gateway switch comprises:

switching said telephone call through a local switch to said PSTN; and

switching said telephone call from said PSTN to said originating VoIP gateway switch.

3. The method of claim 2, wherein said local switch comprises a central office (CO).

4. The method of claim 2, wherein said local switch is selected from the group consisting of a Local Exchange Carrier (LEC), an Incumbent Local Exchange Carrier (ILEC), a Competitive Local Exchange Carrier (CLEC), a Data Local Exchange Carrier (DLEC), a Post, Telegraph and Telephone (PTT), an IntereXchange Carrier (IXC) and an Internet Service Provider (ISP).

5. (Amended) The method of claim 1, wherein said determining said preferred route comprises:

determining a telephone number associated with a calling party, said destination telephone number and any internal identification information about said calling party; and

determining switching parameters for said terminating VoIP gateway switch based on selected routing criteria.

6. The method of claim 5, wherein said determining said telephone number associated with said calling party comprises automatic number identification.

7. The method of claim 5, wherein said determining said telephone number associated with said calling party comprises calling line identification.

8. The method of claim 5, wherein said selected routing criteria comprises least-cost routing.

9. The method of claim 5, wherein said selected routing criteria comprises quality of service.

10. The method of claim 5, wherein said selected routing criteria comprises grade of service.

11. The method of claim 5, wherein said selected routing criteria comprises preferred carrier.

12. (Amended) The method of claim 1, wherein said setting up said two-way communication comprises:

sending an enhanced SS7 signaling initiation packet comprising:

said destination telephone number;

an originating port address of a VoIP module in said originating VoIP gateway switch for receiving voice packets from said terminating VoIP gateway switch; and

a list of available vocoders for voice compression and decompression;

selecting a terminating port address of a VoIP module in said terminating VoIP gateway switch for receiving said voice packets from said originating VoIP gateway switch;

selecting a vocoder from said available vocoder list for voice compression and decompression to be used at said originating and said terminating VoIP gateway switches; and

returning an enhanced SS7 signaling reply packet comprising:

said terminating port addresses; and

said selected vocoder.

13. (Amended) The method of claim 12, wherein said selecting said terminating port address further comprises:

identifying available circuit-switched network trunk groups connected to said terminating VoIP gateway switch having switching circuits available for terminating said telephone call to said destination through said PSTN in accordance with selected routing criteria;

selecting a switching circuit configured for connection to one of said identified available circuit-switched network trunk groups; and

identifying said terminating port address of said VoIP module associated with said selected switching circuit in said terminating VoIP gateway switch having available VoIP capacity.

14. The method of claim 12, wherein said selected routing criteria comprises any combination of least-cost routing, quality of service, grade of service and preferred carrier.

15. The method of claim 1, wherein said originating VoIP gateway switch and said terminating VoIP gateway switch each comprise an STX or IPAX compatible gateway switch.

16. The method of claim 15, wherein said STX or IPAX compatible gateway switch comprises a plurality of T1/E1 circuit boards.

17. (Amended) The method of claim 16, wherein each of said plurality of T1/E1 circuit boards comprises:

T1/E1 connection circuitry configured for switching conventional voice signals to and from said PSTN;

a local time slot interchanger (TSI) connected to said T1/E1 connection circuitry;

a backplane TSI in communication with said local TSI and a pulse code modulated (PCM)/ time division multiplexed (TDM) backplane for interfacing with other of said T1/E1 circuit boards also connected to said PCM/TDM backplane in said STX or IPAX compatible gateway switch; and

a VoIP module connected to said local TSI and configured for sending and receiving packets through said IP-based packet network.

18. (Amended) The method of claim 17, wherein said VoIP module further includes a vocoder for compressing and encoding said voice signals and for decompressing and decoding voice packets.

19. The method of claim 1, further comprising:

communicating voice signals over said IP-based packet network; and

tearing down said telephone call after a calling party or a called party hangs up, disconnects, or terminates said telephone call, or either VoIP gateway switch forces a disconnect or termination of said telephone call for any reason.

20. (Amended) A Voice over Internet Protocol (VoIP) gateway switch for switching VoIP telephone calls over an IP-based packet network comprising:

a pulse code modulated (PCM) and time division multiplexed (TDM) backplane;

a system central processor unit (CPU) board configured to communicate with said IP-based packet network and for controlling said VoIP gateway switch; and

a plurality of T1/E1 circuit boards, each in communication with said system CPU board, each of said plurality of T1/E1 circuit boards comprising:

T1/E1 connection circuitry configured for switching conventional voice signals over a public switched telephone network (PSTN);

a local time slot interchanger (TSI) connected to said T1/E1 connection circuitry;

a backplane TSI in communication with said local TSI and said PCM/TDM backplane for routing said voice signals to and from another of said plurality of T1/E1 circuit boards connected to said PCM/TDM backplane in said VoIP gateway switch; and

a VoIP module connected to said local TSI and configured for sending and receiving packets through said IP-based packet network.

21. (Amended) A Voice over Internet Protocol (VoIP) gateway switch for switching VoIP telephone calls over an IP-based packet network comprising:

a pulse code modulated (PCM) and time division multiplexed (TDM) backplane;

a system central processor unit (CPU) board configured to communicate with said IP-based packet network and for controlling said VoIP gateway switch;

a plurality of T1/E1 circuit boards, each in communication with said system CPU board, each of said plurality of T1/E1 circuit boards comprising:

T1/E1 connection circuitry configured for switching conventional voice signals over a public switched telephone network (PSTN);

a VoIP module configured for sending and receiving packets through said IP-based packet network; and

a local time slot interchanger (TSI) connected to said T1/E1 connection circuitry and said VoIP module for routing said VoIP telephone calls between said PSTN and said IP-based packet network; and

a backplane TSI in communication with said PCM/TDM backplane for routing said voice signals between two of said plurality of T1/E1 circuit boards.

22. (Amended) A system for placing Voice over Internet Protocol (VoIP) telephone calls comprising:

an originating telephone;

a destination telephone;

a local switch connected to said originating telephone through conventional analog or digital telephone lines for switching a telephone call originating between said originating telephone and a public switched telephone network (PSTN);

an originating VoIP gateway switch in communication with said PSTN and in communication with an IP-based packet network for transmitting packets, said packets comprising:

enhanced SS7 signaling packets for setting up and tearing down said VoIP telephone calls; and

voice packets for carrying voice data over said IP-based packet network;

a terminating VoIP gateway switch in communication with said PSTN and in communication with said IP-based packet network and configured for receiving and sending said packets over said IP-based packet network and transmitting voice signals over said PSTN; and

a remote switch for switching said voice signals between said terminating VoIP gateway switch and said destination telephone over said PSTN and said conventional analog or digital telephone lines.

23. The system of claim 22, wherein said IP-based packet network comprises an Internet.

24. The system of claim 23, wherein said Internet comprises a private Internet including bandwidth on demand.

25. A method for setting up Voice over Internet Protocol (VoIP) telephone calls using voice packets over an IP-based packet network or a public switched telephone network (PSTN) comprising:

providing an originating VoIP gateway switch configured to communicate over said IP-based packet network and said PSTN;

providing a terminating VoIP gateway switch configured to communicate over said IP-based packet network and said PSTN;

initiating a VoIP telephone call through said originating VoIP gateway switch;

sending an enhanced SS7 signaling initiate packet from said originating VoIP gateway switch to said terminating VoIP gateway switch over said IP-based packet network;

determining a preferred route for completing said VoIP telephone call;

setting up terminating VoIP settings based on said determined preferred route;

sending an enhanced SS7 signaling reply packet back to said originating VoIP gateway switch including said terminating VoIP settings;

receiving said enhanced SS7 signaling reply packet and finalizing said originating VoIP settings;

sending an enhanced SS7 signaling handshake packet and voice packets to start said VoIP telephone call;

exchanging said voice packets until a terminating event occurs; and

tearing down said VoIP telephone call by exchanging an optionally enhanced SS7 signaling terminate packet.

26. The method of claim 25, wherein said enhanced SS7 signaling initiate packet comprises:

destination telephone number;

a port address in said originating VoIP gateway switch for receiving said voice packets from said terminating VoIP gateway switch; and

a vocoder list in said originating VoIP gateway switch for compressing and decompressing voice signals sent and received, respectively, by said originating VoIP gateway switch over said IP-based packet network.

27. The method of claim 26, wherein said enhanced SS7 signaling reply packet comprises:

a terminating port address in said terminating VoIP gateway switch for receiving said voice packets from said originating VoIP gateway switch; and

a vocoder selected from said vocoder list in said terminating VoIP gateway switch for compressing and decompressing voice signals sent and received, respectively, by said terminating VoIP gateway switch over said IP-based packet network.

28. (Amended) A circuit card for switching voice signals from a public switched telephone network (PSTN) to an IP-based packet network comprising:

T1/E1 connection circuitry configured for switching conventional voice signals to and from said PSTN;

a local time slot interchanger (TSI) connected to said T1/E1 connection circuitry;

a backplane TSI in communication with said local TSI and a pulse code modulated (PCM)/ time division multiplexed (TDM) backplane for interfacing with T1/E1 circuit boards also connected to said PCM/TDM backplane in Specialty Telecommunications Exchange (STX™) or Integrated Protocols and Applications Xchange (IPAX™) compatible gateway switch; and

a Voice over Internet Protocol (VoIP) module connected to said local TSI and configured for sending and receiving packets through said IP-based packet network.

29. (Amended) The circuit card of claim 28, wherein said VoIP module further comprises a vocoder configured for compressing said voice signals, generating packets, receiving said packets and decompressing said packets to generate said voice signals.

30. The circuit card of claim 29, wherein said vocoder further comprises a plurality of selectable voice coding and decoding features to be selected by a terminating VoIP gateway switch on a per call basis.

31. (Amended) A method for providing Voice over Internet Protocol (VoIP) telephone calls over an IP-based packet network comprising:

determining a least-cost routing for a destination telephone number;

selecting an available circuit-switched telephone network trunk having available IP-based packet network switching resources; and

selecting a VoIP module at a terminating gateway based on said least-cost routing and said available IP-based packet network switching resources.

32. The method of claim 31, further comprising exchanging enhanced SS7 signaling packets over said IP-based packet network.

33. (Amended) A method for increasing capacity of a Voice over Internet Protocol (VoIP) gateway switch comprising providing a localized time slot interchanger (TSI) on a T1/E1 circuit card including a VoIP module for communication over an IP-based packet network for on-board routing of a call between said IP-based packet network and a public switched telephone network (PSTN).

34. (Amended) A method for reducing call setup time between an originating Voice over Internet Protocol (VoIP) gateway switch and a terminating VoIP gateway switch comprising exchanging enhanced SS7 signaling packets between said originating VoIP gateway switch and said terminating VoIP gateway switch to provide for least-cost, look-ahead routing of VoIP telephone calls.

35. (Amended) A method of reducing cost of setting up a Voice over Internet Protocol (VoIP) telephone call comprising exchanging enhanced SS7 signaling packets between an originating VoIP gateway switch and a terminating VoIP gateway switch to provide for least-cost, look-ahead routing at said terminating VoIP gateway switch.